IN THE CLAIMS:

1. (original) Method for enhancing communication in a noisy environment comprising:

receiving input signals emanating from at least two microphone arrays each comprising at least two microphones, processing the input signals of each microphone array by a beamformer to determine temporal and spatial information about the input signals of each microphone array.

2. (original) Method according to claim 1, wherein processing the input signals of each microphone array comprises processing by a wanted signal beamformer to obtain a wanted signal and by a blocking beamformer to obtain a blocking signal, preferably wherein the wanted signal beamformer is an adaptive beamformer.

3. (original) Method according to claim 2, wherein processing the input signals of each microphone array further comprises deciding whether a signal is trans- mitted from a wanted signal direction, wherein the wanted signal beamformer is an adaptive beamformer being adapted only if no signal is transmitted from the wanted signal direction.

4. (original) Method according to claim 3, wherein deciding comprises determining a wanted signal power and a blocking signal power, wherein the wanted signal beamformer is adapted only if the blocking signal power is larger than a predetermined constant times the wanted signal power.

5. (original) Method according to one of the preceding claims, further comprising detecting speech activity for each microphone array.

6. (original) Method according to claim 5, wherein detecting speech activity for a microphone array comprises:

determining a wanted signal power, a blocking signal power, and a background noise signal power, comparing the wanted signal power with the blocking signal power and the background noise signal power.

7. (original) Method according to claim 6, further comprising comparing the wanted signal powers of at least two microphone arrays and determining a highest power.

8. (original) Method according to one of the claims 5-7, further comprising applying an attenuation to the processed input signals of a microphone array if no speech activity is detected for the microphone array.

9. (original) Method according to claim 8, wherein applying the attenuation is per- formed adaptivety, preferably by varying the attenuation in predetermined time steps between zero attenuation and a predetermined maximum attenuation.

10. (original) Method according to one of the preceding claims, wherein processing comprises adaptive determining a gain control of the input signals for each microphone array.

11. (original) Method according to claim 10, wherein determining a gain control is performed adaptively.

12. (original) Method according to one of the preceding claims, further comprising selecting at least one output channel out of at least two output channels on which the processed signals are to be output.

13. (original) Method according to claim 12, wherein selecting the at least one output channel comprises determining an amplification for each selected output channel.

14. (original) Computer program product directly loadable into an internal memory of a digital computer, comprising software code portions for performing the steps of the method according to one of the claims 1 to 13.

15. (original) Computer program product stored on a medium readable by a computer system, comprising computer readable program means for causing a computer to perform the steps of the method according to one of the claims 1 to 13.

16. (original) Communication system comprising:

at least two microphone arrays each comprising at least two microphones to produce microphone signals,

at least one analog/digital converter having an input for receiving said microphone signals and an output for providing digital microphone signals,

digital signal processing means having an input for receiving the digital microphone signals, being configured to process the digital microphone signals of each microphone array by a beamformer to determine temporal and spatial information about the microphone signals of each microphone array, and having an output to provide processed output signals to at least two loudspeakers.

17. (original) Communication system according to claim 16, wherein the digital signal processing means is further configured to detect speech activity for each microphone array.

18. (original) Communication system according to claim 17, wherein the digital signal processing means is further configured to determine and apply an attenuation to the processed digital microphone signals of a microphone array if no speech activity is detected for the microphone array.

19. (original) Communication system according to one of the claims 16-18, wherein the digital signal processing means is further configured to select at least one loudspeaker out of the at least two loudspeakers on which the processed signals are to be output.

20. (original) Vehicular cabin comprising a communication system according to one of the claims 16-19 and at least two loudspeakers, wherein each microphone array and each loudspeaker is associated with a passenger seat.

21. (new) Communication system comprising:

multiple microphone arrays each comprising multiple microphones to produce signals corresponding to aural content;

at least one analog/digital converter having an input for receiving said signals and an output for providing respective digital signals; and

digital signal processing means having an input for receiving the digital signals, being configured to process the digital signals of each microphone array to determine temporal and spatial information about the signals of each microphone array, and having an output to provide processed output signals to at least two loudspeakers.